



Congestion Control for Mixed-Loss Environments

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Packet Loss and Delay Bound TCP's Performance

- TCP assumes that all loss is congestion,
 - cuts its transmission rate in half any time that data loss occurs during a round trip time
 - transmission rate rebuilds slowly
- Mathis, et al show the effects of loss on TCP's Congestion Avoidance algorithm:
- When packet loss is not due to congestion, e.g., due to error or channel outage, the result is a limit on TCP's rate that is independent of channel rate (too low)

$$BW < \left(\frac{MSS}{RTT}\right)\frac{1}{\sqrt{p}}$$

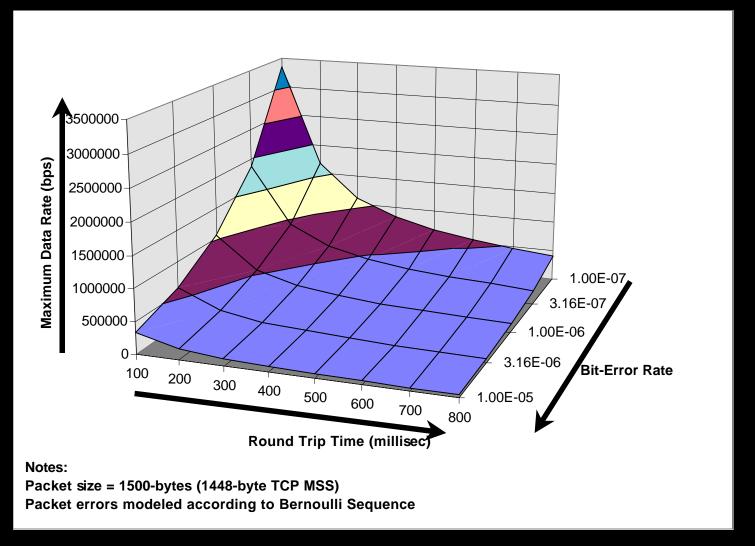
BW = bandwidth

MSS = maximum TCP segment size

RTT = Round Trip Time

p = packet loss probability

Loss and Delay Limit TCP's Maximum Sustainable Rate



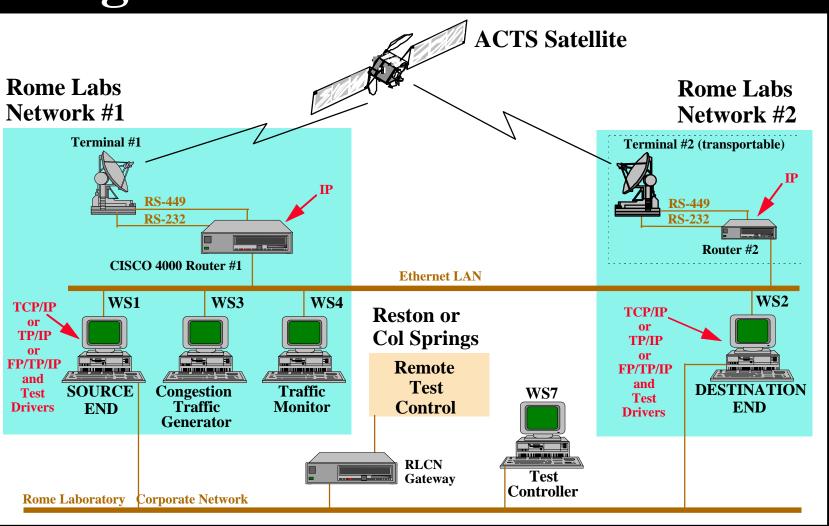
Our goal: Effective Congestion Control in the Presence of Errors

- Much of our work relates to the satellite environment, where RTTs are large (so recovery is slow) and links may be errorprone
- As higher-rate RF-based media become integrated with the Internet, effects of non-congestion loss will become more pronounced
- We are attempting to develop congestion control mechanisms that are effective, fair, and that do not use loss as an indication of congestion
- Similar and related work
 - TCP-Vegas measures changes in throughput and infers the amount of queuing in the network
 - Packet-Pair Attempts to determine bottleneck data rate by examining packet spacing

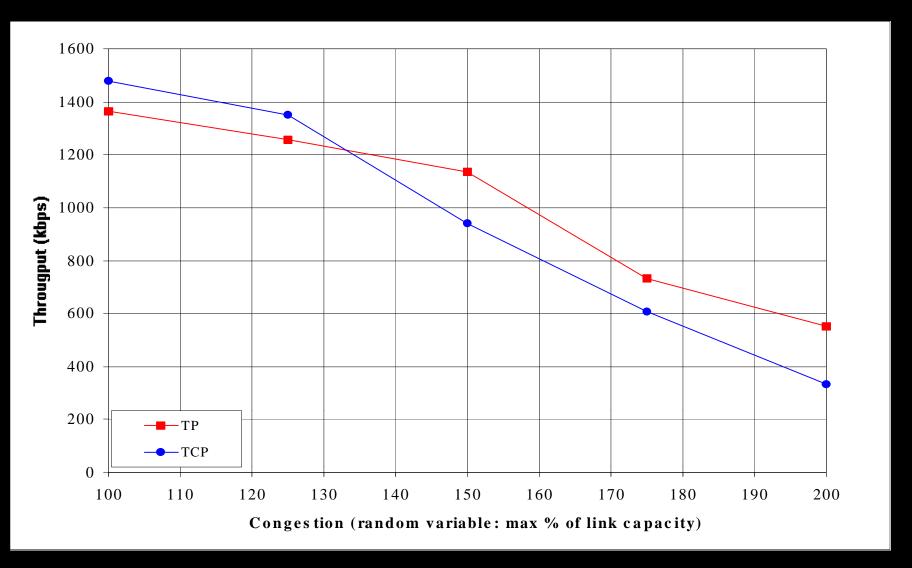
TCP-Vegas and Modifications

- We used TCP-Vegas as the basis for extension
 - Two modes of operation:
 - Exponential similar to TCP's slow start, but slower
 - Linear analogous to TCP's congestion avoidance
 - In it's linear mode, Vegas can increase or decrease congestion window by 1 packet/RTT or leave it unchanged (TCP can only increase)
 - Still reduces congestion window when loss is detected
- Our initial modifications:
 - Do not reduce congestion window upon loss
 - Add Explicit Congestion Notification from the network (to avoid problems with starting up in a congested environment)
 - Integrated rate control with congestion control

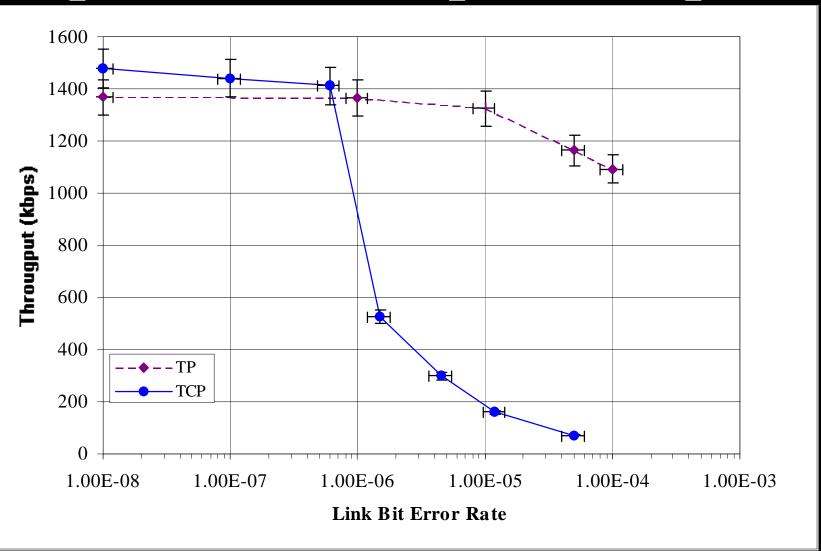
Comparative Results: Test Configuration



Comparative Congestion Response



Comparative Corruption Response



Problems with this approach

- TCP-Vegas's slow start is too slow (doubles the transmission rate every *other* round trip time, but still can overrun available bandwidth)
- With (very) high bandwidth-delay product environments, need to reduce congestion window more quickly than 1 packet/RTT
- Uses round-trip times to calculate *data* throughput -- "noise" on the ACK channel corrupts calculation
- Becomes completely confused by large changes in propagation delays

Alternatives

- Fix our modified Vegas
 - Use a different slow start algorithm
 - Revise algorithm to reduce congestion window proportional to queueing detected
 - Synchronize sender and receiver clocks to eliminate ACK channel timing effects
 - Still doesn't handle changing propagation delays
- Use a Receiver-Based Packet-Pair or Packet Bunch Mode-based approach to measure available bandwidth
 - Still must synchronize clocks
 - May want to integrate with TCP-Vegas to "damp" changes

Conclusions

- We believe that it is possible to develop a congestion control mechanism that works effectively but does not depend on loss as an indication of congestion
- Our revisions to TCP-Vegas are roughly comparable to TCP's congestion control response in the presence of congestion, and superior to TCP's in the presence of corruption
 - Problems remain with the Vegas-based approach
 - May require use of an alternative capacity measurement mechanism, such as Packet-Pair or Packet Bunch Mode
 - These approaches require synchronization of sender and receiver clocks to allow focus on *data* channel
- Deployment will be challenging